Patent Claims

1. A method of reducing echo and/or noise signals in telecommunications systems for transmitting useful acoustic signals, particularly human speech, comprising determining by silence detection when the mixture of useful signals and interference signals contains a speech signal or when a silence interval is present, and varying, by means of a two-input multiplier, the amplitude of the useful signals, which are generally disturbed by echo and/or noise signals, in response to a time-dependent control signal $a_0(t)$ or a control signal $a_0(k)$ clocked at a sampling rate $f_T=1/T$, where $k \in \aleph$ denotes the number of samples, and T denotes the period from one sample to the next,

   characterized in

   that the control signal $a_0(t)$ or $a_0(k)$ is varied in such a way that in the presence of speech signals in the useful signal, the amplitude of the control signal $a_0(t)$ or $a_0(k)$ is set to a predetermined constant value c_0, that from the beginning of a silence interval in the useful signal, the amplitude of the control signal $a_0(t)$ or $a_0(k)$ is continuously reduced from one sample to the next according to the recursion formula

   $$a_0(k+1) = a_0(k) \cdot \beta, \qquad \text{where } \beta < 1,$$

   and that after the end of a silence interval, $a_0(k)$ is set equal to c_0.

2. A method as claimed in claim 1, characterized in that the factor β is determined from the sampling rate f_T, a time constant τ_1, and a predefined constant factor c_1 according to the relation

$$\beta = c_1 \cdot \exp(-1/\tau_1 f_T).$$

3. A method as claimed in claim 2, characterized in that the time constant τ_1 is chosen to be between 50 ms and 150 ms, preferably $\tau_1 \approx 65$ ms.

4. A method as claimed in any one of the preceding claims, characterized in that the constant value c_0 is chosen to be equal to 1.

5. A method as claimed in any one of the preceding claims, characterized in that during a silence interval and/or in the presence of an echo signal $a_0(k+1)$ assumes a predefined constant value c_2 if the preceding value $a_0(k)$ has become less than or equal to c_2.

6. A method as claimed in any one of claims1 to 4, characterized in that during a silence interval and/or in the presence of an echo signal and for $a_0(k) \leq c_2$, where c_2 is a predefined constant, the power value of the noise level N in the communications channel currently being used is continuously measured and/or estimated, and that depending on the current noise level N, the control signal $a_0(k+1)$ is continuously adjusted according to $a_0(k+1) = f(N)$, where f(N) is a predetermined function of N.

7. A method as claimed in claim 6, characterized in that the predetermined function f(N) is a function g(S/N), which depends on the quotient S/N of the power value of the signal level S of the useful signals to be transmitted and the power value of the noise level N, or that the predetermined function f(N) is a function g'(N/S), which depends on the reciprocal of said quotient.

8. A method as claimed in claim 7, characterized in that, if $1/N << 1$ or $S/N = 0$ dB, the function f(N) or g(S/N) begins with a constant value $f_0 > 0$ or $g_0 > 0$, respectively, rises to a maximum f_{max} or g_{max} in the range between N or S/N = 10 dB to 15 dB, respectively,

preferably at N or S/N $\approx$ 12 dB, respectively, and then decreases to a minimum value f_{min} or g_{min}, respectively, preferably to 0 dB, where 5 dB $\leq f_0, g_0 \leq$ 10 dB, preferably 6 dB $\leq f_0, g_0 \leq$ 8 dB, and where 20 dB $\leq f_{max}, g_{max} \leq$ 30 dB, preferably $f_{max}, g_{max} \approx$25 dB.

9. A method as claimed in any one of claims 6 to 8, characterized in that the function f(N) or g(S/N) is linear at least in sections, preferably in all its sections.

10. A method as claimed in any one of claims 6 to 8, characterized in that the function f(N) or g(S/N) consists of polynomials and is a skewed bell-shaped curve.

11. A method as claimed in any one of claims 6 to 10, characterized in that the functions f(N) and g(S/N) or g'(N/S) are chosen such that the reduction of the noise level N is aurally compensated in accordance with the psychoacoustic mean value of the spectrum audible by the human ear.

12. A method as claimed in any one of the preceding claims, characterized in that in addition to the detection and reduction of noise signals, the presence of echo signals is detected and/or predicted, and that the echo signals are suppressed or reduced.

13. A method as claimed in claim 12 and in any one of claims 6 to 11, characterized in that the control signal $a_0(k+1)$ is continuously adjusted according to $a_0(k+1) = h(N, S, ES, \tau_E, ERL)$, where $h(N, S, ES, \tau_E, ERL)$ is a predetermined function of the noise level N, the signal level S, the useful signal ES in the opposite direction from a speaking party, the constant delay τ_E of the echo signal, and an attenuation constant ERL of the amplitude of the echo signal.

14. A method as claimed in claim 12, characterized in that the reduction of noise signals and the reduction of echo signals are controlled separately.

15. A method as claimed in any one of claims 12 to 14, characterized in that during the time of an echo reduction, an artificial noise signal is added to the useful signal.

16. A method as claimed in claim 15, characterized in that the artificial noise signal comprises an acoustic signal sequence perceived to be psychoacoustically pleasant (= comfort noise).

17. A method as claimed in claim 15, characterized in that the artificial noise signal comprises a noise signal previously recorded during the current communication.

18. A method as claimed in any one of the preceding claims, characterized in
that in a silence detector (SPD), a short-time output signal sam(x), a medium-time output signal mam(x), and a long-time output signal lam(x) are formed by means of a short-time level estimator, a medium-time level estimator, and a long-time level estimator, respectively,
that the three output signals sam(x), mam(x), and lam(x) are so adjusted via suitable amplification coefficients that they are approximately equal in magnitude when the input signal x is a pure noise signal, with
$sam(x) < mam(x) < lam(x)$,
that the three output signals sam(x), mam(x), and lam(x) are monitored by comparators, and
that the presence of a speech signal as the input signal x is assumed when both sam(x) and mam(x) first become larger than lam(x), while the presence of a silence interval is assumed when thereafter sam(x) and/or mam(x) become smaller than lam(x).

19. A method as claimed in claim 18, characterized in that for silence interval estimation, the three output signals sam(x), mam(x), and lam(x) are fed to a neural network which was trained with a plurality of scenarios with different input signals x.

20. A method as claimed in any one of the preceding claims, characterized in that the useful signal to be transmitted is subjected to a spectral subtraction.

21. A method as claimed in any one of the preceding claims, characterized in that the useful signal to be transmitted is subjected to spectral filtering adapted to the sense of human hearing.

22. A server unit for supporting the method claimed in any one of claims 1 to 21.

23. A computer program for carrying out the method claimed in any one of claims 1 to 21.